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EXAMINER	
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/550,230	VIEILLEDENT ET AL.	
	Examiner	**Art Unit**	
	GEORGE C. MONIKANG	2614	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE _3_ MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on _15 September 2008_.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) _1-30_ is/are pending in the application.

4a) Of the above claim(s) _1,5,9 and 10_ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) _2-4,6-8 and 11-30_ is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☒ Certified copies of the priority documents have been received in Application No. 10/550,230.

3.☒ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Response to Arguments

1. Applicant's arguments, filed 9/15/2008, with respect to the rejection(s) of claim(s) 2-4, 6-8, 11-30 under 10/550,230 have been fully considered and are persuasive. Therefore, the rejection has been withdrawn. However, upon further consideration, a new ground(s) of rejection is made in view of Cooper et al, US Patent 5,333,200.

2. Newly added claim 30 is analyzed and rejected as claim 1.

Claim Rejections - 35 USC § 102

1. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

2. Claims 1, 11-14 & 29 are rejected under 35 U.S.C. 102(b) as being anticipated by Cooper et al, US Patent 5,333,200.

Re Claim 1, Cooper et al discloses a method for processing an electric sound signal wherein a right sound signal and a left sound signal are diffused in a reflective environment by two speakers (*Cooper et al, col. 23, lines 6-22*) and are detected by an acoustic detector comprising a right microphone and a left microphone (*Cooper et al, fig. 1b: 143 & 145*), the method comprising: computing a first temporal filter corresponding to a detection by the right microphone of the right sound signal (*Cooper et al, fig. 1b: 132, 134, 136, 138: is inherent that a filter can calculate the transfer*

function); computing a second temporal filter corresponding to a detection by the left microphone of the right sound signal (_Cooper et al, fig. 1b: 132, 134, 136, 138: is inherent that a filter can calculate the transfer function_); computing a third temporal filter corresponding to a detection by the left microphone of the left sound signal _Cooper et al, fig. 1b: 132, 134, 136, 138: is inherent that a filter can calculate the transfer function_); computing a fourth temporal filter corresponding to a detection by the right microphone of the left sound signal (_Cooper et al, fig. 1b: 132, 134, 136, 138: is inherent that a filter can calculate the transfer function_); modifying each of the temporal filters by an operation including at least one of: normalizing the temporal filters on a maximum of a direct field or on a quadratic average, temporal resetting of the temporal filters in relation to each other, providing a time lag of samples from a temporal filter (_Cooper et al, col. 9, lines 37-49_), masking of at least some of the samples from the temporal filter, and altering an amplitude of at least some of the samples from a temporal filter (_Cooper et al, col. 19, line 59 through col. 20, line 14_); applying the modified temporal filters to a right original sound signal and a left original sound signal to obtain processed electric sound signals by: applying a first modified temporal filter to the right original electric sound signal to obtain a first processed electric sound signal (_Cooper et al, fig. 1b: 132, 134, 136, 138: is inherent that a filter can calculate the transfer function_), applying a second modified temporal filter to the right original electric sound signal to obtain a second processed electric sound signal (_Cooper et al, fig. 1b: 132, 134, 136, 138: is inherent that a filter can calculate the transfer function_), applying a third modified temporal filter to the left original sound signal to obtain a third processed electric sound

signal (*Cooper et al, fig. 1b: 132, 134, 136, 138: is inherent that a filter can calculate the transfer function*), and applying a fourth modified temporal filter to the left original sound signal to obtain a fourth processed electric sound signal (*Cooper et al, fig. 1b: 132, 134, 136, 138: is inherent that a filter can calculate the transfer function*), adding the first and fourth processed electric sound signals and the right original sound signal to obtain a right processed electric sound signal (*Cooper et al, fig. 1b: 126 & 128*); adding the second and third processed electric sound signals and the left original sound signal to obtain a left processed electric sound signal (*Cooper et al, fig. 1b: 126 & 128*); and diffusing the right processed electric sound signal and the left processed sound signal (*Cooper et al, fig. 1b*).

Re Claim 11, Cooper et al disclose the method according to claim 1 wherein combined electric sound signals on the right and left are filtered on given frequency bands (*Cooper et al, col. 9, lines 37-68*) and, a delay is introduced in each of these frequency bands (*Cooper et al, col. 9, lines 37-68*).

Re Claim 12, Cooper et al disclose the method according to claim 11, wherein combined electric sound signals on the right and left are filtered by using a high-pass filter (*Cooper et al, col. 14, lines 6-29*), and high-frequency electric sound signals are obtained, combined electric sound signals on the right and left are filtered by using a low-pass filter (*Cooper et al, col. 13, lines 8-21*), and low-frequency electric sound signals are obtained.

Re Claim 13, Cooper et al disclose the method according to claim 12, wherein a first delay is introduced in the low-frequency electric sound signals (*Cooper et al, col. 9,*

lines 37-68) and a second delay is introduced in the high-frequency electric sound signals (*Cooper et al, col. 9, lines 37-68*).

Re Claim 14, Cooper et al disclose the method according to claim 13, wherein the first delay introduced in the low-frequency electric sound signal obtained from the combined electric sound signal on the right is different from the first delay introduced in the low-frequency electric sound signal obtained from the combined electric sound signal on the left (*Cooper et al, col. 9, lines 37-68*), and the second delay introduced in the high-frequency electric sound signal obtained from the combined electric sound signal on the right is different from the second delay introduced in the high-frequency electric sound signal obtained from the combined electric sound signal on the left (*Cooper et al, col. 9, lines 37-68*).

Re Claim 29, Cooper et al disclose the method according to claim 1, wherein a time lag is introduced between the original electric sound signal and the processed electric sound signals (*Cooper et al, col. 9, lines 37-49*).

Claim Rejections - 35 USC § 103

3. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4. The factual inquiries set forth in *Graham* **v.** *John Deere Co.*, 383 U.S. 1, 148 USPQ 459 (1966), that are applied for establishing a background for determining obviousness under 35 U.S.C. 103(a) are summarized as follows:

1. Determining the scope and contents of the prior art.
2. Ascertaining the differences between the prior art and the claims at issue.
3. Resolving the level of ordinary skill in the pertinent art.
4. Considering objective evidence present in the application indicating obviousness or nonobviousness.

5. Claims 2, 7-8 are rejected under 35 U.S.C. 103(a) as being unpatentable over Cooper et al, US Patent 5,333,200 as applied to claim 1 above, in view of Ishii, US Patent 6,961,433 B2.

Re Claim 2, the combined teachings of Cooper et al discloses the method according to claim 1, but fails to disclose wherein the simulating includes: producing a white acoustic sound signal on the right is with an acoustic diffusion system, from a white noise electric signal (*Ishii, fig. 6: 6R1; col. 10, lines 46-61*); detecting with an acoustic detector a corresponding acoustic signal received in the form of a modified white received electric sound signal on the right and a modified white electric sound signal on the left corresponding to the reception of the white acoustic sound signal on the right (*Ishii, fig. 6: BR1; col. 10, lines 46-61*); producing a frequency spectrum on the right corresponding to a white noise electric signal on the right, and two received frequency spectrums, respectively corresponding to the modified white received electric sound signal on the right and to the modified white received electric sound signal on the left (*Ishii, fig. 6: 6L1; col. 10, lines 46-61)*; producing a first set of coefficients from frequency filters from the frequency spectrum on the right and from the frequency

spectrum of the modified white received electric sound signal on the right (*Ishii, fig. 8: FR1R2; col. 12, line 61 through col. 13, line 12*); producing a second set of coefficients from frequency filters from the frequency spectrum on the right and from the frequency spectrum of the modified white received electric sound signal on the left (*Ishii, fig. 8: FL1R2; col. 12, line 61 through col. 13, line 12*); producing a white acoustic sound signal on the left with an acoustic diffusion system, from a white noise electric signal (*Ishii, col. 10, lines 46-61*); detecting a corresponding acoustic signal received in the form of a modified white received electric sound signal on the left and a modified white electric sound signal on the right corresponding to the reception of the white acoustic sound signal on the left with an acoustic detector (*Ishii, fig. 6: BR1, BR2, BL1, BL2; col. 10, lines 46-61*); producing a frequency spectrum on the left corresponding to a white noise electric signal on the left (*Ishii, fig. 6: GL1L2; col. 10, lines 46-61*), and two received frequency spectrums, respectively corresponding to the modified white received electric sound signal on the left and to the modified white received electric sound signal on the right (*Ishii, fig. 6: GL1, GR1; col. 10, lines 46-61*); producing a third set of coefficients from frequency filters from the frequency spectrum on the left and from the frequency spectrum of the modified white received electric sound signal on the left (*Ishii, fig. 8: FL1L2; col. 12, line 61 through col. 13, line 12*); producing a fourth set of coefficients from frequency filters from the frequency spectrum on the left and from the frequency spectrum of the modified white received electric sound signal on the right (*Ishii, fig. 8: FR1L1; col. 12, line 61 through col. 13, line 12*), said four sets of coefficients forming a quadrille of coefficient sets (*Ishii, col. 12, line 61 through col. 13, line 12: 22R1, 22R2,*

_22L1, 22L2_); and filtering the electric sound signals on the right and left with frequency filters whose parameters are given by said quadrille (_Ishii, col. 12, line 61 through col. 13, line 12: 22R1, 22R2, 22L1, 22L2_). However, Ishii does. It would have been obvious for Ishii to use the white acoustic sound signal with the method of processing electrical signals of Cooper et al for the purpose of obtaining sound depth.

Re Claim 7, the combined teachings of Cooper et al and Ishii disclose the method according to claim 2 wherein quadrilles of sets of coefficients are produced for different configurations of the acoustic diffusion system and or for different rooms in which the acoustic diffusion system is placed for the production of coefficients (_Ishii, col. 10, lines 46-61_).

Re Claim 8, the combined teachings of Cooper et al and Ishii disclose the method according to claim 7, wherein one of the configurations is a configuration in cone of confusion (_Ishii, fig. 1: user's ears_).

Claim 3 is rejected under 35 U.S.C. 103(a) as being unpatentable over Cooper et al, US Patent 5,333,200 and Ishii, US Patent 6,961,433 B2 as applied to claim 2 above, in view of Breebaart et al, US Patent 7,181,019 B2.

Re Claim 3, Cooper et al and Ishii disclose the method according to claim 2, but fail to disclose wherein: the sets of coefficients are produced from the two spectrums by a component to component complex division of complex points from these components in each of these spectrums as taught in Breebaart et al (_Breebaart et al, col. 5, lines 26-34_). It would have been obvious to use the complex division of Breebaart et al with the

method of processing electrical signals of Cooper et al for the purpose of calculating the phase difference.

Claims 4-6 are rejected under 35 U.S.C. 103(a) as being unpatentable over Cooper et al, US Patent 5,333,200 and Ishii, US Patent 6,961,433 B2, as applied to claim 2 above, in view of Ueno et al, US Patent 5,960,390.

Re Claim 4, the combined teachings of Cooper et al and Ishii disclose the method according to claim 2, but fail to disclose wherein said diffusion includes the steps of producing the coefficients from four temporal filters from coefficients of the first, second, third and fourth frequency filters respectively as taught in Ueno et al (*Ueno et al, fig. 5: 104a-104d; col. 10, lines 13-24*) for the purpose of converting the signals to time domain.

Re Claim 5, the combined teachings of Cooper et al, Ishii and Ueno et al disclose the method according to claim 4, wherein the coefficients of temporal filters are modified by an operation including at least one of the steps of: normalizing temporal filters of a quadrille, on the maximum of the direct field or on quadratic average of the diffuse field (*Ueno et al, fig. 5: 105a-105d*); temporal resetting of the temporal filters with relation to each other (*Ueno et al, fig. 5: 104a-104d; col. 10, lines 13-24*); providing a time lag of samples from a temporal filter; masking of some samples from the temporal filter (*Ueno et al, fig. 5: 104a-104d; col. 10, lines 13-24*); alteration of amplitudes from certain samples from a temporal filter (*Ueno et al, col. 5, line 66 through col. 6, line 14*).

Re Claim 6, the combined teachings of Cooper et al, Ishii and Ueno et al disclose the method according to claim 4 wherein the coefficients from a temporal filter those whose rank is greater than a given rank are eliminated and where in the coefficients from a temporal filter those whose value is lower than a threshold are eliminated (*Ueno et al, col. 12, lines 15-28*).

Claims 15-17, 19 & 21-28 are rejected under 35 U.S.C. 103(a) as being unpatentable over Cooper et al, US Patent 5,333,200 as applied to claim 1 above, in view of Ueno et al, US Patent 5,960,390.

Re Claim 15, Cooper et al disclose the method according to claim 1, where the filtering coefficients are coefficients of finite impulse response filters (*Cooper et al, col. 9, lines 37-68*), but fails to disclose characterized in that, wherein, to filter, a signal transform of an electric sound signal is performed and a transformed signal is obtained, the transformed signal is multiplied by the filtering coefficients and a multiplied signal is obtained, the multiplied signal is transformed by an inverse transform as taught in Ueno et al (*Ueno et al, fig. 5: 104a-104d, 107; col. 13, lines 17-25*). It would have been obvious to use the signal transformation of Ueno et al with the method to process electrical signals of Cooper et al for the purpose of effectively preventing pre-echo and post-echo from being generated and can perform effective coding to which an psycho-acoustic model is applied.

Re Claim 16, the combined teachings of Cooper et al and Ueno et al disclose the method according to claim 15, wherein, to perform the transform a frame of the electric

sound symbol is divided into N blocks (*Ueno et al, fig. 5: 101*), the transform of each of the blocks is performed (*Ueno et al, fig. 5: 104a-104d*), the filtering coefficients are divided into N packets of coefficients (*Ueno et al, fig. 5: 101*), the N blocks of input data are multiplied two by two by the N packets of filter coefficients (*Ueno et al, fig. 5: 107; col. 11, lines 38-44*), and the multiplied blocks are added to obtain the multiplied signal (*Ueno et al, fig. 5: 107; col. 11, lines 38-44*).

Re Claim 17, the combined teachings of Cooper et al and Ueno et al disclose the method according to claim 16, wherein to divide the frame and to calculate the transform (*Ueno et al, fig. 5: 101, 104a-104d*), the transform of each of the N blocks is calculated successively (*Ueno et al, fig. 5: 101, 104a-104d*), and the transformed blocks are transmitted to a delay line at N outputs (*Cooper et al, col. 9, lines 37-68*).

Re Claim 19, which further recites “wherein, to divide a frame of the signal into N blocks, double blocks are formed that are overlayed on each other by half, the transform of each of the double blocks is performed, the N packets of coefficients are completed by the constant samples to obtain double packets, each of the N double blocks are multiplied by one of the N double packets and multiplied double blocks are obtained, and the multiplied blocks are extracted from the multiplied double blocks.” Cooper et al and Ueno et al do not explicitly disclose the above limitations as claimed. Official notice is taken that both the concept and advantages of the above limitations are well known in the art. It would have been obvious to divide a frame of the signal into N blocks, double blocks are formed that are overlayed on each other by half, the transform of each of the double blocks is performed, the N packets of coefficients are completed by the constant

samples to obtain double packets, each of the N double blocks are multiplied by one of the N double packets and multiplied double blocks are obtained, and the multiplied blocks are extracted from the multiplied double blocks since the blocks are divided within a circular buffer.

Re Claim 21, the combined teachings of Cooper et al and Ueno et al disclose the method according to claim 1 wherein, to diffuse, equalization functions are incorporated in the cells situated upstream from the Fourier transform cells (*Ueno et al, col. 2, lines 1-16*).

Re Claim 22, the combined teachings of Cooper et al and Ueno et al the method according to claim 21, wherein the frequency components of four frequency filters obtained from four modified temporal filters are adjusted independently (*Ueno et al, fig 5: 104a-104d*).

Re Claim 23, the combined teachings of Cooper et al and Ueno et al disclose the method according to claim 1 wherein, to diffuse, the phase and/or the amplitude of the temporal filter coefficients are modified along all or part of the impulse response (*Ueno et al, col. 5, line 66 through col. 6, line 14; fig. 5:104a-104d*).

Re Claim 24, which further recites "wherein, to perform the transform the filtering temporal coefficients are divided into Q slots (HDD1-HDD4) of coefficients with progressive length M, 2M, 4M, . . . (2 (Q-1))M points, the transform of each of these slots is performed and transformed slots are obtained, a frame of the electric sound signal is divided into blocks (x1-x8) with a length of M points, the transform of each of these blocks is performed and transformed blocks are obtained, and the transformed

blocks are multiplied by the transformed slots and corresponding multiplied blocks are obtained by inverse transformation to the blocks of signals that half-overlap each other two by two in time." Cooper et al and Ueno et al do not explicitly disclose the above limitations as claimed. Official notice is taken that both the concept and advantages of the above limitations are well known in the art. It would have been obvious to perform the transform the filtering temporal coefficients are divided into Q slots (HDD1-HDD4) of coefficients with progressive length M, 2M, 4M, . . . (2 (Q-1))M points, the transform of each of these slots is performed and transformed slots are obtained, a frame of the electric sound signal is divided into blocks (x1-x8) with a length of M points, the transform of each of these blocks is performed and transformed blocks are obtained, and the transformed blocks are multiplied by the transformed slots and corresponding multiplied blocks are obtained by inverse transformation to the blocks of signals that half-overlap each other two by two in time since the transformations are discrete Fourier transforms and inverse discrete Fourier transforms.

Claims 25-28 have been analyzed and rejected according to claim 24.

Claim 18 is rejected under 35 U.S.C. 103(a) as being unpatentable over Cooper et al, US Patent 5,333,200 and Ueno et al, US Patent 5,960,390 as applied to claim 16 above, in view of Parry et al, US Patent 6,535,920 B1.

Re Claim 18, the combined teachings of Cooper et al and Ueno et al disclose the method according to claim 16 wherein, to divide the frame into N blocks (*Ueno et al, fig. 5: 101*), but fail to disclose an electric sound signal is stored in a circular buffer memory

with capacity proportional to the nth of the frame of the electric sound signal as taught in Parry et al (*Parry et al, fig. 7: 124*) for the purpose of storing processed signals.

Claim 20 is rejected under 35 U.S.C. 103(a) as being unpatentable over Cooper et al, US Patent 5,333,200 as applied to claim 1, in view of Ishii, US Patent 6,961,433 B2, and further in view of Ueno et al, US Patent 5,960,390.

Re Claim 20, Cooper et al disclose the method according to claim 1 but fail to disclose wherein, to simulate, an artificial head that comprises two acoustic detectors are placed in a median axis of two acoustic diffusion systems (*Ishii, fig. 6: BR1, BR2, BL1, BL2; col. 10, lines 46-61*), direct fields and crossed fields received by the acoustic detectors are aligned two by two by varying the position of the artificial head (*Ishii, fig. 6: BR1, BR2, BL1, BL2; col. 10, lines 46-61*). It would have been obvious to use the varying positions of the artificial head of Ishii with the method of processing electrical signals of Cooper et al for the purpose of obtaining sound depth. Cooper et al and Ishii also fail to disclose an electric signal in the form of a Dirac comb is applied simultaneously as input to the two acoustic diffusion systems as taught in Ueno et al (*Ueno et al, fig. 5: 104a-104d; col. 10, lines 13-24)* for the purpose of modifying sound or original sound recordings in order to give the listener optimal listening comfort.

Contact

Any inquiry concerning this communication or earlier communications from the examiner should be directed to GEORGE C. MONIKANG whose telephone number is

(571)270-1190. The examiner can normally be reached on M-F. alt Fri. Off 7:30am-5:00pm (est).

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Chin Vivian can be reached on 571-272-7848. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/George C Monikang/
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12/12/2008

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